UNITED STATES PATENT AND TRADEMARK OFFICE

UNITED STATES DEPARTMENT OF COMMERCE
United States Patent and Trademark Office
Address: COMMISSIONER FOR PATENTS
P.O. Box 1450
Alexandria, Virginia 22313-1450
www.uspto.gov

APPLICATION NO.	FILING DATE	FIRST NAMED INVENTOR	ATTORNEY DOCKET NO.	CONFIRMATION NO.
10/701,865	11/05/2003	Joseph J. Kubler	14364US03	7803

7590 02/03/2011

Christopher C. Winslade
McAndrews, Held & Malloy, Ltd
34th Floor
500 W. Madison St.
Chicago, IL 60661

EXAMINER	
ANWAR, MOHAMMAD S	
ART UNIT	**PAPER NUMBER**
2463	

MAIL DATE	DELIVERY MODE
02/03/2011	PAPER

Please find below and/or attached an Office communication concerning this application or proceeding.

The time period for reply, if any, is set in the attached communication.

Office Action Summary	Application No. 10/701,865	Applicant(s) KUBLER ET AL.	
	Examiner MOHAMMAD ANWAR	**Art Unit** 2463	

-- The MAILING DATE of this communication appears on the cover sheet with the correspondence address --

Period for Reply

A SHORTENED STATUTORY PERIOD FOR REPLY IS SET TO EXPIRE *3* MONTH(S) OR THIRTY (30) DAYS, WHICHEVER IS LONGER, FROM THE MAILING DATE OF THIS COMMUNICATION.

- Extensions of time may be available under the provisions of 37 CFR 1.136(a). In no event, however, may a reply be timely filed after SIX (6) MONTHS from the mailing date of this communication.
- If NO period for reply is specified above, the maximum statutory period will apply and will expire SIX (6) MONTHS from the mailing date of this communication.
- Failure to reply within the set or extended period for reply will, by statute, cause the application to become ABANDONED (35 U.S.C. § 133). Any reply received by the Office later than three months after the mailing date of this communication, even if timely filed, may reduce any earned patent term adjustment. See 37 CFR 1.704(b).

Status

1)☒ Responsive to communication(s) filed on *17 November 2010*.

2a)☒ This action is **FINAL**. 2b)☒ This action is non-final.

3)☐ Since this application is in condition for allowance except for formal matters, prosecution as to the merits is closed in accordance with the practice under *Ex parte Quayle*, 1935 C.D. 11, 453 O.G. 213.

Disposition of Claims

4)☒ Claim(s) *22-81* is/are pending in the application.

4a) Of the above claim(s) _____ is/are withdrawn from consideration.

5)☐ Claim(s) _____ is/are allowed.

6)☒ Claim(s) *22-81* is/are rejected.

7)☐ Claim(s) _____ is/are objected to.

8)☐ Claim(s) _____ are subject to restriction and/or election requirement.

Application Papers

9)☐ The specification is objected to by the Examiner.

10)☐ The drawing(s) filed on _____ is/are: a)☐ accepted or b)☐ objected to by the Examiner.

Applicant may not request that any objection to the drawing(s) be held in abeyance. See 37 CFR 1.85(a).

Replacement drawing sheet(s) including the correction is required if the drawing(s) is objected to. See 37 CFR 1.121(d).

11)☐ The oath or declaration is objected to by the Examiner. Note the attached Office Action or form PTO-152.

Priority under 35 U.S.C. § 119

12)☐ Acknowledgment is made of a claim for foreign priority under 35 U.S.C. § 119(a)-(d) or (f).

a)☐ All b)☐ Some * c)☐ None of:

1.☐ Certified copies of the priority documents have been received.

2.☐ Certified copies of the priority documents have been received in Application No. _____.

3.☐ Copies of the certified copies of the priority documents have been received in this National Stage application from the International Bureau (PCT Rule 17.2(a)).

* See the attached detailed Office action for a list of the certified copies not received.

Attachment(s)

1) ☒ Notice of References Cited (PTO-892)

2) ☐ Notice of Draftsperson's Patent Drawing Review (PTO-948)

3) ☐ Information Disclosure Statement(s) (PTO/SB/08) Paper No(s)/Mail Date _____.

4) ☐ Interview Summary (PTO-413) Paper No(s)/Mail Date. _____ .

5) ☐ Notice of Informal Patent Application

6) ☐ Other: _____.

DETAILED ACTION

Response to Arguments

1. Applicant's arguments filed 11/17/10 have been fully considered but they are not persuasive. Please see response below:

In response to applicant argument, the Office asserts that Fuller teaches "call is branched out to PSTN or packet." However, the portion of Fuller shown above, specifically chosen by the Office, makes no mention of a "packet," or of a "packet-based network," or of "a user selected one of a circuit switched network and a packet-based network," let alone Applicants' claimed "database for use in
voice call routing to cause delivery of voice to a called party by a user selected one of a circuit switched network and a packet-based network," as recited by claim 22. Further, the Office has not identified where Fuller supports the assertion by the Office that a "call is branched out to PSTN or packet [network]." The cited portions of Fuller do not provide the support required by M.P.E.P. §2142 for such an assertion. (Please see newly cited reference Bonnaure et al. disclose packet/circuit switching user defined calls).

Claim Rejections - 35 USC § 103

1. The following is a quotation of 35 U.S.C. 103(a) which forms the basis for all obviousness rejections set forth in this Office action:

> (a) A patent may not be obtained though the invention is not identically disclosed or described as set forth in section 102 of this title, if the differences between the subject matter sought to be patented and the prior art are such that the subject matter as a whole would have been obvious at the time the invention was made to a person having ordinary skill in the art to which said subject matter pertains. Patentability shall not be negatived by the manner in which the invention was made.

2. Claim 22, 25, 26, 28, 29, 32-34, 36, 39, 40, 41, 47, 50, and 57-59 are rejected under 35 U.S.C. 103(a) as being unpatentable over Berken (WO 91/08629) in view of Richter (US006104706A) and Bonnaure et al. (U.S. Patent No. 5,862,339).

Regarding Claims 22, 28, 29, 36 and 47, Berken discloses a system for processing voice for communication over a network (see FIG. 1A, wireless telecommunication system for voice and data communication; see page 4, line 6-9), the system comprising:

conversion circuitry (see FIG. 1C, phone interface 209) for converting analog voice signals to digital voice data (see FIG. 1C, phone interface 209 converts sound/voice input from telephone 127 into digital voice packets; see page 6, line 16-20) and for converting digital voice data to analog voice signals for the reproduction of voice (see FIG. 1C, phone interface 209 converts received digitized voice packets back into analog/sound signals for the telephone 127; see page 5, line 28 to page 6, line 5);

a processing circuit (see FIG. 1C, a combined system of processor 215, switch 213, phone 209) for managing the packetization of digital voice data to provide digital voice data packets (see FIG. 1C, a combined system 215,213,209 controls/manages converting of voice data to digital voice packets; see page 6, line 5-20) and for managing the depacketization of digital voice data (see FIG. 1C, a combined system 215,213,209 controls/manages converting of received digitized voice packets back into analog/sound signals for the telephone 127; see page 5, line 28 to page 6, line 5), the processing circuit packetizing the digital voice data according to a packet protocol (see

FIG. 1C, a combined system 215,213,209 converting voice data in accordance with packet protocol/rule for transmission; see page 6, line 16-20); and

a transceiver circuit for wireless transmission and wireless reception (see FIG. 1A, C, Radio interface 211 circuitry/module which perform both transmitter and receiver functionalities) according to a wireless communication protocol of the digital voice data packets (FIG. 1C, see page 6, line 14-20; radio interface 211 of a user module 103 communicates by utilizing packet protocol/practice/procedure/rules), wherein the digital voice data packets comprises information (see FIG. 3, control time slot of frame; and/or FIG. 4, packet header of the voice time slot) used for routing the digital voice data packets (see page 9, line 1-10; see page 10, line 17-30; control time slot of the transmit/receive frame comprises control information for routing/forwarding through PSTN, Ethernet LAN, or Token Ring LAN; and/or a packet header of the voice time slot comprises control information routing/forwarding through PSTN, Ethernet LAN, or Token Ring LAN);

<u>a database</u> (see FIG. 1C, memory 217; see page 6, line 5-9; see page 7, line 15-19), <u>and the use in voice call routing to cause delivery of voice to a called party</u> (see FIG. 5, 6, using control information for routing voice call to the called system; see page 9, line 1-10; see page 10, line 17-30) <u>by a user selected one of a circuit switched network and a packet-based network</u> (see FIG. 1A, 6, by a wireless system's request to select either circuit switch path for voice call to PSTN 151 (i.e. circuit switched network) or a packet switch path to Ethernet LAN (i.e. packet switch network)) <u>according to a</u>

information (see FIG. 5, 6, according to a request; see page 10, lines 25 to col. 11, lines 5; see page 9, lines 15-25).

Although Berken discloses a database and a user selected one of a circuit switched network and a packet-based network as set forth above,

Berken does not explicitly disclose "*a packet is a unit of information transmitted as a whole from one device to another over the network*", "*destination*", "*having at least one entry comprising user defined call routing information and at least one associated destination address*", "*for use in voice call routing to cause delivery of voice to a called party*", and "*according to a destination address of the called party and the database*".

However, a definition of *a packet, which is a unit of information transmitted as a whole from one device to another over the network* is well known in the art per prior art Microsoft computer dictionary (*published since 1990 four years before applicant's invention as admitted by applicant as prior art, see remark page 24, submitted 12/2/09*), and voice packet comprising destination information for routing is so well known in the art so that it would identify and locate the recipient of the voice data packet. In particular, Richter teaches a packet protocol wherein a packet is a unit of information transmitted as a whole from one device to another over the network (see FIG. 6, data packet protocol with a data packet 52, note that according to applicant admitted Microsoft 1992 published dictionary, *a packet, is a unit of information transmitted as a whole from one device to another over the network;* see col. 6, line 60 to col. 7, line 20);

wherein the digital voice data packets comprise destination information used for routing (see FIG. 6, destination address 76, max destination count 74, active destination

count 72, and destination count that used for routing; see col. 6, line 60 to col. 7, line 20) the digital voice packets through the communication network (see FIG. 5, for routing voice packets over the network between two callers; see col. 5, line 36-66; col. 6, line 44-56);

comparing a destination address to a database (see FIG. 8-10, looking-up/comparing the destination address in the table lookup (i.e. FIG. 8, Table Lookup 98; see FIG. 9, Table Lookup 818; see FIG. 10, Table lookup 922, 925) having at least one entry comprising user defined call routing information (see FIG. 8, 9, 10, machine address and stream address in the table lookup) and at least one associated destination address (see FIG. 8-10, destination address (e.g. 924 (e.g. 2D) per FIG. 10)), the database for use in voice call routing to cause delivery of voice to a called party (see FIG. 8-10, Table lookup is used in audio call routing to routed audio to caller 2 (see FIG. 4); see col. 5, line 52-60; see col. 6, line) by a user selected one of a packet-based network or circuit switch network (see FIG. 4, by a caller 1 selection one of a Ethernet (i.e. packet switch network) or telephone line/network (i.e. circuit switch network); see col. 11, line 50-65) according to a destination address of the called party and the database (see col. 12, line 5-53; according to destination address of the caller 2 and a table lookup).

Therefore, it would have been obvious to one having ordinary skill in the art at the time the invention was made to provide "*a packet is a unit of information transmitted as a whole from one device to another over the network*", "*destination*", "*having at least one entry comprising user defined call routing information and at least one associated*

destination address", "*for use in voice call routing to cause delivery of voice to a called party*", and "*according to a destination address of the called party and the database*, as taught by Richter and well established teaching in art in the system of Berken, so that it would provide capability to the caller and callee to hear each other; see Richter col. 7, line 10-19, and it would also identify and locate the recipient of the voice data packet. Berken and Richter disclose all the subject matter but fails to mention explicitly a database having at least one entry comprising user defined call routing information and at least one associated destination address, the database for use in voice call routing to cause delivery of voice to a called party by a user selected one of a circuit switched network and a packet-based network according to a destination address of the called party and the database. However, Bonnaure et al. from a similar field of endeavor disclose a database having at least one entry comprising user defined call routing information (see Figure 17 (1734); customer criteria list) and at least one associated destination address, the database for use in voice call routing to cause delivery of voice to a called party by a user selected one of a circuit switched network and a packet-based network according to a destination address of the called party and the database (see column 3 lines 2-8, column 4 lines 36-42). Thus, it would have been obvious to one ordinary skill in the art at the time of invention was made to include Bonnaure et al. database scheme into Berken and Richter transmission scheme. The method can be implemented in a database. The motivation of doing this is to schedule calls based on user defined parameters.

Regarding Claims 25, 33,40,57,58 and 59, Berken disclose a frequency hopping spread spectrum technique (see page 11, line 20-31; frequency hoping system of spread spectrum coding).

Regarding Claims 26, 34, and 41, Berken disclose a direct sequence spread spectrum technique (see page 11, line 20-31; direct sequence spread spectrum coding).

Regarding Claims 32, 39, 50, Berken discloses conversion circuitry for converting analog voice signals to digital voice data (see FIG. 1C, phone interface 209 converts sound/voice input from telephone 127 into digital voice packets for radio transmission; see page 6, line 16-20) and for converting digital voice data to analog voice signals for the reproduction of voice (see FIG. 1C, phone interface 209 converts digitized voice packets received from radio interface back into analog/sound signals for the telephone 127; see page 5, line 28 to page 6, line 5).

3. Claims 43 and 46 are rejected under 35 U.S.C. 103(a) as being unpatentable over Berken in view of Richter, Bonnaure et al. and further in view of Harrison (US 5,796,727).

Regarding Claim 43, Berken discloses a system for processing voice for communication over a network, the system (see FIG. 1A, wireless telecommunication system for voice communication; see page 4, line 6-9) comprising:

a processing circuit (see FIG. 1C, a combined system of processor 215, switch 213, phone 209) for managing the packetization of digital voice data to provide digital voice data packets (see FIG. 1C, a combined system 215,213,209 controls/manages

converting of voice data to digital voice packets; see page 6, line 6-20) and for managing the depacketization of digital voice data (see FIG. 1C, a combined system 215,213,209 controls/manages converting of received digitized voice packets back into analog/sound signals for the telephone 127; see page 5, line 28 to page 6, line 5), the processing circuit packetizing the digital voice data according to a packet protocol (see FIG. 1C, a combined system 215,213,209 converting voice data in accordance with packet protocol/rule for transmission; see page 6, line 16-20); wherein the digital voice data packets comprises information (see FIG. 3, control time slot of frame; and/or FIG. 4, packet header of the voice time slot) used for routing the digital voice data packets (see page 9, line 1-10; see page 10, line 17-30; control time slot of the transmit/receive frame comprises control information for routing/forwarding through PSTN, Ethernet LAN, or Token Ring LAN; and/or a packet header of the voice time slot comprises control information routing/forwarding through PSTN, Ethernet LAN, or Token Ring LAN);

a transceiver circuit for wireless transmission and wireless reception (see FIG. 1A, C, Radio interface 211 circuitry/module which perform both transmitter and receiver functionalities) according to a wireless communication protocol of the digital voice data packets (FIG. 1C, see page 6, line 14-20; radio interface 211 of a user module 103 communicates by utilizing packet protocol/practice/procedure/rules);

<u>a database</u> (see FIG. 1C, memory 217; see page 6, line 5-9; see page 7, line 15-19), <u>and the use in voice call routing to cause delivery of voice to a called party</u> (see FIG. 5, 6, using control information for routing voice call to the called system; see page

9, line 1-10; see page 10, line 17-30) by a user selected one of a circuit switched network and a packet-based network (see FIG. 1A, 6, by a wireless system's request to select either circuit switch path for voice call to PSTN 151 (i.e. circuit switched network) or a packet switch path to Ethernet LAN (i.e. packet switch network)) according to a information (see FIG. 5, 6, according to a request; see page 10, lines 25 to col. 11, lines 5; see page 9, lines 15-25).

Although Berken discloses a database and a user selected one of a circuit switched network and a packet-based network as set forth above,

Berken does not explicitly disclose "*a packet is a unit of information transmitted as a whole from one device to another over the network*", "*destination*", "*having at least one entry comprising user defined call routing information and at least one associated destination address*", "*for use in voice call routing to cause delivery of voice to a called party*", and "*according to a destination address of the called party and the database*".

However, a definition of *a packet, which is a unit of information transmitted as a whole from one device to another over the network* is well known in the art per prior art Microsoft computer dictionary (*published since 1990 four years before applicant's invention as admitted by applicant as prior art, see remark page 24, submitted 12/2/09*), and voice packet comprising destination information for routing is so well known in the art so that it would identify and locate the recipient of the voice data packet. In particular, Richter teaches a packet protocol wherein a packet is a unit of information transmitted as a whole from one device to another over the network (see FIG. 6, data packet protocol with a data packet 52, note that according to applicant admitted

Microsoft 1992 published dictionary, *a packet, is a unit of information transmitted as a whole from one device to another over the network;* see col. 6, line 60 to col. 7, line 20*)*;

wherein the digital voice data packets comprise destination information used for routing (see FIG. 6, destination address 76, max destination count 74, active destination count 72, and destination count that used for routing; see col. 6, line 60 to col. 7, line 20) the digital voice packets through the communication network (see FIG. 5, for routing voice packets over the network between two callers; see col. 5, line 36-66; col. 6, line 44-56);

comparing a destination address to a database (see FIG. 8-10, looking-up/comparing the destination address in the table lookup (i.e. FIG. 8, Table Lookup 98; see FIG. 9, Table Lookup 818; see FIG. 10, Table lookup 922, 925) having at least one entry comprising user defined call routing information (see FIG. 8, 9, 10, machine address and stream address in the table lookup) and at least one associated destination address (see FIG. 8-10, destination address (e.g. 924 (e.g. 2D) per FIG. 10)), the database for use in voice call routing to cause delivery of voice to a called party (see FIG. 8-10, Table lookup is used in audio call routing to routed audio to caller 2 (see FIG. 4); see col. 5, line 52-60; see col. 6, line) by a user selected one of a packet-based network or circuit switch network (see FIG. 4, by a caller 1 selection one of a Ethernet (i.e. packet switch network) or telephone line/network (i.e. circuit switch network); see col. 11, line 50-65) according to a destination address of the called party and the database (see col. 12, line 5-53; according to destination address of the caller 2 and a table lookup).

Therefore, it would have been obvious to one having ordinary skill in the art at the time the invention was made to provide "*a packet is a unit of information transmitted as a whole from one device to another over the network*", "*destination*", "*having at least one entry comprising user defined call routing information and at least one associated destination address*", "*for use in voice call routing to cause delivery of voice to a called party*", and "*according to a destination address of the called party and the database*, as taught by Berken and well established teaching in art in the system of Berken, so that it would provide capability to the caller and callee to hear each other; see Richter col. 7, line 10-19, and it would also identify and locate the recipient of the voice data packet. Berken and Richter disclose all the subject matter but fails to mention explicitly a database having at least one entry comprising user defined call routing information and at least one associated destination address, the database for use in voice call routing to cause delivery of voice to a called party by a user selected one of a circuit switched network and a packet-based network according to a destination address of the called party and the database. However, Bonnaure et al. from a similar field of endeavor disclose a database having at least one entry comprising user defined call routing information (see Figure 17 (1734); customer criteria list) and at least one associated destination address, the database for use in voice call routing to cause delivery of voice to a called party by a user selected one of a circuit switched network and a packet-based network according to a destination address of the called party and the database (see column 3 lines 2-8, column 4 lines 36-42). Thus, it would have been obvious to one ordinary skill in the art at the time of invention was made to include Bonnaure et al.

database scheme into Berken and Richter transmission scheme. The method can be implemented in a database. The motivation of doing this is to schedule calls based on user defined parameters.

Berken, Richter and Bonnaure et al. fails to explicitly disclose “a media access controller for controlling operation”.

However, Harrison teaches wherein the digital voice packets (see col. 4, line 45-49; 65 to col. 5, line 7; packets of voice data) comprise destination information used for routing the outgoing digital voice packets (see FIG. 5; MS adding destination address into packet for routing through network (see FIG. 1); see col. 6, line 5-12; see col. 7, line 35 to col. 8, line 15; see col.12, line 39 to col. 13, line 11); a media access controller (see col. 5, line 25-31; MAC) for controlling the operation of the transceiver to transmit and receive information according to a wireless communication protocol (see col. 12, line 39-61; MAC controls/process transmit and receive information according to IEEE wireless protocol).

Therefore, it would have been obvious to one having ordinary skill in the art at the time the invention was made to provide “MAC”, as taught by Harrison in the combined system of Berken, Richter and Bonnaure et al. so that it would ensure to establish and route the packets of voice data to destination end user, provide various classes of data communication services as well as voices services, and provide registration and channel/bandwidth allocation; see Harrison col. 3, line 22-26; see col. 4, line 50-55; see col. 7, line 35-55.

Regarding Claim 46, Berken discloses conversion circuitry for converting analog voice signals to digital voice data (see FIG. 1C, phone interface 209 converts sound/voice input from telephone 127 into digital voice packets for radio transmission; see page 6, line 16-20) and for converting digital voice data to analog voice signals for the reproduction of voice (see FIG. 1C, phone interface 209 converts digitized voice packets received from radio interface back into analog/sound signals for the telephone 127; see page 5, line 28 to page 6, line 5).

1. Claims 27, 35 and 42 rejected under 35 U.S.C. 103(a) as being unpatentable over Berken in view of Richter and Bonnaure et al. as applied to claims 22, 29 and 36 above, and further in view of Weaver (US005956673A).

Regarding Claims 27, 35, 42, Berken discloses wireless transmission and reception of digital voice data packets/transceiver utilizes a communication protocol (see FIG. 1A, controls/manage a radio transmission according to a radio protocol (i.e. TDMA); see page 10, line 23-33 for voice packet in PSTN or data packet in Ethernet LAN, or Token Ring LAN; see page 6, line 5 to page 8, line 4) that accommodates a plurality of bandwidth (see page 10, line 4 to col. 11, line 15; radio protocol provides different bandwidth for different services/data type).

Neither Berken nor Richter explicitly discloses “data rates including at least a standard data rate and a higher data rate”.

However, Weaver discloses a processor (see FIG. 1, Encoder 180) for controlling the operation of the radio according to a communication protocol that accommodates a

plurality of data rates (see col. 1, line 25-37; see col. 5, line 55-59; see col. 9, line 33-34; plurality of data rates) including at least a standard data rate and a higher data rate (see col. 1, line 25-37; see col. 6, line 13-25; see col. 9, line 33-35; low or less than full (i.e. half or quarter) data rate and full data rate).

Therefore, it would have been obvious to one having ordinary skill in the art at the time the invention was made to provide data rates including at least a standard data rate and a higher data rate, as taught by Weaver in the combined system of Berken, Richter and Bonnaure et al., so that it would provide avoid the disadvantage of tandem vocoding; see Weaver col. 1, line 60-67.

4. Claims 51 and 54 are rejected under 35 U.S.C. 103(a) as being unpatentable over Berken in view of Richter and Bonnaure et al., and further in view of Weaver (US005956673A).

Regarding Claim 51, Berken discloses a system for processing voice for communication over a network (see FIG. 1A, wireless telecommunication system for voice communication; see page 4, line 6-9) comprising:

a processing circuit (see FIG. 1C, a combined system of processor 215, switch 213, phone 209) for managing the packetization of digital voice data to provide digital voice data packets (see FIG. 1C, a combined system 215,213,209 controls/manages converting of voice data to digital voice packets; see page 6, line 6-20) and for managing the depacketization of digital voice data (see FIG. 1C, a combined system 215,213,209 controls/manages converting of received digitized voice packets back into

analog/sound signals for the telephone 127; see page 5, line 28 to page 6, line 5), wherein the digital voice data packets comprises destination information (see FIG. 3, control time slot of frame; and/or FIG. 4, packet header of the voice time slot) used for routing the digital voice data packets (see page 9, line 1-10; see page 10, line 17-30; control time slot of the transmit/receive frame comprises routing/forwarding information through PSTN, Ethernet LAN, or Token Ring LAN; and/or a packet header of the voice time slot comprises routing/forwarding information through PSTN, Ethernet LAN, or Token Ring LAN), the processing circuit packetizing the digital voice data according to a packet protocol (see FIG. 1C, a combined system 215,213,209 converting voice data in accordance with packet protocol/rule for transmission; see page 6, line 16-20); and

a radio for wireless transmission and reception (see FIG. 1A, C, Radio interface 211 circuitry/module which perform both transmitter and receiver functionalities) of digital voice data packets (FIG. 1C, see page 6, line 14-20; radio interface 211 of a user module 103 communicates by utilizing packet protocol/practice/procedure/rules) and

a processor (see FIG. 1C, processor 215) for controlling the operation of the radio according to a communication protocol (see FIG. 1A, controls/manage a radio transmission according to a radio protocol (i.e. TDMA); see page 10, line 23-33 for voice packet in PSTN or data packet in Ethernet LAN, or Token Ring LAN; see page 6, line 5 to page 8, line 4) that accommodates a plurality of bandwidth (see page 10, line 4 to col. 11, line 15; radio protocol provides different bandwidth for different services/data type).

Berken does not explicitly disclose “destination”.

However, voice packet comprising destination information for routing is so well known in the art so that it would identify and locate the recipient of the voice data packet. In particular, Richter teaches wherein the digital voice data packets comprise destination information used for routing (see FIG. 6, destination address 76, max destination count 74, active destination count 72, and destination count that used for routing; see col. 6, line 60 to col. 7, line 20) the digital voice packets through the communication network (see FIG. 5, for routing voice packets over the network between two callers; see col. 5, line 36-66; col. 6, line 44-56).

Therefore, it would have been obvious to one having ordinary skill in the art at the time the invention was made to provide destination, as taught by Berken and well established teaching in art in the system of Berken, so that it would provide capability to the caller and callee to hear each other; see Richter col. 7, line 10-19, and it would also identify and locate the recipient of the voice data packet.

Berken, Richter and Bonnaure et al. fails to explicitly disclose “data rates including at least a standard data rate and a higher data rate”.

Weaver discloses a processor (see FIG. 1, Encoder 180) for controlling the operation of the radio according to a communication protocol that accommodates a plurality of data rates (see col. 1, line 25-37; see col. 5, line 55-59; see col. 9, line 33-34; plurality of data rates) including at least a standard data rate and a higher data rate (see col. 1, line 25-37; see col. 6, line 13-25; see col. 9, line 33-35; low or less than full (i.e. half or quarter) data rate and full data rate).

Therefore, it would have been obvious to one having ordinary skill in the art at the time the invention was made to provide data rates including at least a standard data rate and a higher data rate, as taught by Weaver in the combined system of Berken and Richter, so that it would provide avoid the disadvantage of tandem vocoding; see Weaver col. 1, line 60-67.

Regarding Claim 54, Berken discloses conversion circuitry for converting analog voice signals to digital voice data (see FIG. 1C, phone interface 209 converts sound/voice input from telephone 127 into digital voice packets for radio transmission; see page 6, line 16-20) and for converting digital voice data to analog voice signals for the reproduction of voice (see FIG. 1C, phone interface 209 converts digitized voice packets received from radio interface back into analog/sound signals for the telephone 127; see page 5, line 28 to page 6, line 5).

2. Claims 23, 24, 30, 31, 37, 38, 48 and 49 are rejected under 35 U.S.C. 103(a) as being unpatentable over Berken in view of Richter and Bonnaure et al. as applied to claims 22, 29, 36 and 47 above, and further in view of Perkins (US005159592A).

Regarding Claims 23, 24,30,31,37,38,48,49, neither Berken nor Richter and Bonnaure et al. explicitly discloses an Internet Protocol (IP), wherein IP protocol is TCP/IP. However, Perkins discloses wherein the wireless packet network uses an Internet Protocol (IP), wherein IP protocol is TCP/IP (see col. 4, line 10-20; see col. 7, line 35-56; col. 8, line 30-45; mobile unit 10 and access gateway utilizing TCP/IP).

Therefore, it would have been obvious to one having ordinary skill in the art at the time the invention was made to provide TCP/IP, as taught by Perkins, in the system of Berken, so that it would provide wireless migration users to a network operating in accordance with the TCP/IP protocol; see Perkins col. 2, line 55-60; see col. 3, line 15-30.

3. Claims 44 and 45 rejected under 35 U.S.C. 103(a) as being unpatentable over Berken in view of Richter and Bonnaure et al. as applied to claim 43 above, and further in view of Perkins (US005159592A).

Regarding Claims 44 and 45, neither Berken, Richter, Bonnaure et al. nor Harrison explicitly disclose an Internet Protocol (IP), wherein IP protocol is TCP/IP. However, Perkins discloses wherein the wireless packet network uses an Internet Protocol (IP), wherein IP protocol is TCP/IP (see col. 4, line 10-20; see col. 7, line 35-56; col. 8, line 30-45; mobile unit 10 and access gateway utilizing TCP/IP).

Therefore, it would have been obvious to one having ordinary skill in the art at the time the invention was made to provide TCP/IP, as taught by Perkins, in the combined system of Berken, Richter, Bonnaure et al. and Harrison, so that it would provide wireless migration users to a network operating in accordance with the TCP/IP protocol; see Perkins col. 2, line 55-60; see col. 3, line 15-30.

4. Claims 52 and 53 are rejected under 35 U.S.C. 103(a) as being unpatentable over Berken in view of Richter and Bonnaure et al. as applied to claim 51 above, and further in view of Perkins (US005159592A).

Regarding Claims 52 and 53, neither Berken, Richter, Bonnaure et al. nor Weaver explicitly discloses an Internet Protocol (IP), wherein IP protocol is TCP/IP. However, Perkins discloses wherein the wireless packet network uses an Internet Protocol (IP), wherein IP protocol is TCP/IP (see col. 4, line 10-20; see col. 7, line 35-56; col. 8, line 30-45; mobile unit 10 and access gateway utilizing TCP/IP).

Therefore, it would have been obvious to one having ordinary skill in the art at the time the invention was made to provide TCP/IP, as taught by Perkins, in the combined system of Berken, Richter and Weaver, so that it would provide wireless migration users to a network operating in accordance with the TCP/IP protocol; see Perkins col. 2, line 55-60; see col. 3, line 15-30.

5. Claims 55 and 56 are rejected under 35 U.S.C. 103(a) as being unpatentable over Berken, Richter and Bonnaure et al. as applied to claim 47 above, and further in view of Cripps (US005838730A).

Regarding Claims 55 and 56, Berken disclose a frequency hopping spread spectrum technique (see page 11, line 20-31; frequency hoping system of spread spectrum coding).

Berken does not explicitly disclose a frequency of approximately 2.4 gigahertz.

However, using 2.4 GHz frequency hopping is well known in the art as defined by FCC. In particular, Cripps discloses wherein the wireless packet network communicates at a frequency of approximately 2.4 gigahertz (abstract; see col. 2, line 13-20; see col. 36, line 32-45; 2.4 GHz).

Therefore, it would have been obvious to one having ordinary skill in the art at the time the invention was made to provide 2.4 GHz, as taught by Cripps, in the combined system of Berken, Bonnaure et al. and Richter, so that it would provide a transmitter/receiver in accordance with FCC rules for 2.4 GHz ISM which is low cost and low power; see Cripps col. 2, line 15-32.

6. Claims 74,75, 76, 77 and 79 are rejected under 35 U.S.C. 103(a) as being unpatentable over Berken in view of Richter and Bonnaure et al. as applied to claims 22, 28, 29, 36 and 47 above, and further in view of Dezonno (US 5991394).

Regarding Claims 74, 75, 76, 77 and 79, Berken discloses a user (see FIG. 1C, wireless user) selecting delivery of voice to called party by one of the circuit switch network and the packet-based network (see FIG. 1A, 6, request to select either circuit switch path for voice call to PSTN 151 (i.e. circuit switched network) or a packet switch path to Ethernet LAN (i.e. packet switch network); see page 10, lines 25 to col. 11, lines 5; see page 9, lines 15-25). Richter also discloses a user (see FIG. 4, caller 1) selecting delivery of voice to called party by one of the circuit switch network and the packet-based network (see FIG. 4, by a caller 1 selection one of a Ethernet (i.e. packet switch network) or telephone line/network (i.e. circuit switch network); see col. 11, line

50-65), indicated by a user defined parameter (see col. 12, line 5-53; according to destination address of the caller 2 and a table lookup, which are defined by the user).

Although Bonnaure et al. disclose a user selecting deliver of voice to the called party by one of the circuit switched network and the packet-based network, indicated by a user defined parameter,

neither Berken nor Richter and Bonnaure explicitly discloses a user is "*prompted*".

However, Dezonno discloses a user (see FIG. 1, Agent 104, or user 102) is prompted to select delivery of voice (see FIG. 1, 2, prompt to select voice telephone call) to the called party (see FIG. 1, to user 102, or agent 104) by <u>one</u> of the circuit switched network (see FIG. 1, by telephone network 122) and the packet-based network (see FIG. 1, by Internet 108), if such prompting is indicated by a user defined parameter (see FIG. 1, when prompting is indicated by user's parameter such as telephone number or name; see col. 3, line 33 to col. 4, line 25; see col. 4, line 40 to col. 5, lines 65; col. 7, line 25-32).

Therefore, it would have been obvious to one having ordinary skill in the art at the time the invention was made to provide a user is "*prompted*" as taught by Dezonno, in the combined system of Berken and Richter, so that it would provide establishing voice communications between a computer user and an agent of a business, and a computer user to easily and conveniently have a business advertising on a computer network, call the computer user back over the telephone; see Dezonno col. 2, line 10-14, col. 3, lines 1-5.

7. Claim 78 is rejected under 35 U.S.C. 103(a) as being unpatentable over Berken in view of Richter, Bonnaure et al. and Harrison as applied to claim 43 above, and further in view of Dezonno (US 5991394).

Regarding Claim 78, Berken discloses a user (see FIG. 1C, wireless user) selecting delivery of voice to called party by one of the circuit switch network and the packet-based network (see FIG. 1A, 6, request to select either circuit switch path for voice call to PSTN 151 (i.e. circuit switched network) or a packet switch path to Ethernet LAN (i.e. packet switch network); see page 10, lines 25 to col. 11, lines 5; see page 9, lines 15-25). Richter also discloses a user (see FIG. 4, caller 1) selecting delivery of voice to called party by one of the circuit switch network and the packet-based network (see FIG. 4, by a caller 1 selection one of a Ethernet (i.e. packet switch network) or telephone line/network (i.e. circuit switch network); see col. 11, line 50-65), indicated by a user defined parameter (see col. 12, line 5-53; according to destination address of the caller 2 and a table lookup, which are defined by the user).

Although Bonnaure et al. disclose a user selecting deliver of voice to the called party by one of the circuit switched network and the packet-based network, indicated by a user defined parameter,

neither Berken, Richter, Bonnaure et al. nor Harrison explicitly discloses a user is "*prompted*".

However, Dezonno discloses a user (see FIG. 1, Agent 104, or user 102) is prompted to select delivery of voice (see FIG. 1, 2, prompt to select voice telephone

call) to the called party (see FIG. 1, to user 102, or agent 104) by <u>one</u> of the circuit switched network (see FIG. 1, by telephone network 122) and the packet-based network (see FIG. 1, by Internet 108), if such prompting is indicated by a user defined parameter (see FIG. 1, when prompting is indicated by user's parameter such as telephone number or name; see col. 3, line 33 to col. 4, line 25; see col. 4, line 40 to col. 5, lines 65; col. 7, line 25-32).

Therefore, it would have been obvious to one having ordinary skill in the art at the time the invention was made to provide a user is "*prompted*" as taught by Dezonno, in the combined system of Berken, Richter, Bonnaure et al. and Harrison, so that it would provide establishing voice communications between a computer user and an agent of a business, and a computer user to easily and conveniently have a business advertising on a computer network, call the computer user back over the telephone; see Dezonno col. 2, line 10-14, col. 3, lines 1-5.

8. Claim 80 is rejected under 35 U.S.C. 103(a) as being unpatentable over Berken in view of Richter, Bonnaure et al. and Weaver as applied to claim 51 above, and further in view of Dezonno (US 5991394).

Regarding Claim 80, Berken discloses a user (see FIG. 1C, wireless user) selecting delivery of voice to called party by <u>one</u> of the circuit switch network and the packet-based network (see FIG. 1A, 6, request to select either circuit switch path for voice call to PSTN 151 (i.e. circuit switched network) or a packet switch path to Ethernet LAN (i.e. packet switch network); see page 10, lines 25 to col. 11, lines 5; see page 9,

lines 15-25). Richter also discloses a user (see FIG. 4, caller 1) selecting delivery of voice to called party by one of the circuit switch network and the packet-based network (see FIG. 4, by a caller 1 selection one of a Ethernet (i.e. packet switch network) or telephone line/network (i.e. circuit switch network); see col. 11, line 50-65), indicated by a user defined parameter (see col. 12, line 5-53; according to destination address of the caller 2 and a table lookup, which are defined by the user).

Although Bonnaure et al. disclose a user selecting deliver of voice to the called party by one of the circuit switched network and the packet-based network, indicated by a user defined parameter,

neither Berken, Richter, Bonnaure et al. nor Weaver explicitly discloses a user is "*prompted*".

However, Dezonno discloses a user (see FIG. 1, Agent 104, or user 102) is prompted to select delivery of voice (see FIG. 1, 2, prompt to select voice telephone call) to the called party (see FIG. 1, to user 102, or agent 104) by <u>one</u> of the circuit switched network (see FIG. 1, by telephone network 122) and the packet-based network (see FIG. 1, by Internet 108), if such prompting is indicated by a user defined parameter (see FIG. 1, when prompting is indicated by user's parameter such as telephone number or name; see col. 3, line 33 to col. 4, line 25; see col. 4, line 40 to col. 5, lines 65; col. 7, line 25-32).

Therefore, it would have been obvious to one having ordinary skill in the art at the time the invention was made to provide a user is "*prompted*" as taught by Dezonno, in the combined system of Berken, Richter, Bonnaure et al. and Weaver, so that it

would provide establishing voice communications between a computer user and an agent of a business, and a computer user to easily and conveniently have a business advertising on a computer network, call the computer user back over the telephone; see Dezonno col. 2, line 10-14, col. 3, lines 1-5.

9. Claims 60, 61, 62, and 68-73 are rejected under 35 U.S.C. 103(a) as being unpatentable over Berken in view of Hutton (US006108704A), and further in view of Reimer (U.S. 4,704,696).

Regarding Claim 60, Berken discloses one or more circuits for use in a handheld communication device supporting the exchange of voice over a communication network (see FIG. 1A, C, circuits/modules/components of wireless user device for voice communication in a network; see page 4, line 6-9), the one or more circuits comprising: at least one interface to circuitry for transmitting and receiving over a radio frequency channel (see FIG. 1A, C, Radio interface 211 circuitry/module for both transmitting and receiving over an RF channel 107; see page 6, line 14-20; page 7, line 25-32), packets comprising packetized digital voice data packetized according to a packet protocol (see FIG. 1 C, packets comprises packetized/converted voice data in accordance with packet protocol/rule for transmission; see page 6, line 16-20); at least one processor (see FIG. 1C, a combined system of processor 215, switch 213, phone 209) operably coupled to the at least one interface (see FIG. 1 C, couples to radio interface 211), the at least one processor operating to, at least, convert analog voice signals at a first user location (see FIG. 1A, first User device; see FIG. 5, first user

module UM1; see page 9, line 28-33) to first digital voice data (see FIG. 1C, phone interface 209 converts sound/voice input from telephone 127 into digital voice data for packetizing; see page 6, line 16-20); packetize the first digital voice data according to the packet protocol to produce first digital voice data packets (see FIG. 1 C, phone interface 209 converts/packetize digital voice data into voice packets; see page 6, line 16-20), wherein the first digital voice data packets comprise information (see FIG. 3, control time slot of frame; and/or FIG. 4, packet header of the voice time slot) used for routing the first digital voice data packets through the communication network (see page 9, line 1-10; see page 10, line 17-30; control time slot of the transmit/receive flame comprises information for routing/forwarding through PSTN, Ethernet LAN, or Token Ring LAN; and/or a packet header of the voice time slot comprises information for routing/forwarding through PSTN, Ethemet LAN, or Token Ring LAN); wirelessly transmit, in accordance with a wireless communication protocol, the first digital voice data packets (see FIG. 1A,C, see page 6, line 14-20; the user module 103 transmits voice packets over radio channel 107 in accordance with radio protocol/practice/procedure/rule); wirelessly receive, in accordance with the wireless communication protocol, second digital voice data packets (see FIG. 1A,C, see page 6, line 14-20; the user module 103 received voice packets from RF channel 107 in accordance with a radio protocol/practice/procedure/rule); depacketize the second digital voice data packets to produce second digital voice data (see FIG. 1C, phone interface 209 depacketizes/converts digitized voice packets back into digitized voice data for the telephone 127; see page 5, line 28 to page 6, line 5); and convert the

second digital voice data to analog voice signals at the location of the first user (see FIG. 1 C, phone interface 209 converts digitized voice data back into analog/sound signals for the telephone 127; see page 5, line 28 to page 6, line 5). Berken does not explicitly disclose "destination and to a second user". However, a user device sending voice packet to another user over the network is well known in the art. In particular, Hutton teaches the first digital voice data packets (see col. 3, line 55-61; see col. 4, line 19-25,65 to col. 5, line 20; see col. 8, line 20-26; IP packet with compressed voice/audio data) comprise destination information (see FIG. 5-6, destination/callee IP address or phone number of second processing unit 22) used for routing the first digital voice data packets (see FIG. 3-4, destination IP address is used for routing the compressed audio data IP packets) through the communication network (see FIG. 3-4, routing though Internet 24) to a second user (see FIG. 3-4, to the remote user/callee user device at second processing unit 22); see col. 5, line 1-65; see col. 7, line 10-35; see col. 8, line 15-45; see col. 10, line 25-60). Therefore, it would have been obvious to one having ordinary skill in the art at the time the invention was made to provide destination and a second user, as taught by Hutton in the system of Berken, so that it would provide exchanging realtime voice/video IP packet with IP address between two end units via Internet; see Hutton col. 1, line 50-65; also by utilization destination information, it enable the caller to route the voice packets to the callee. Neither Berken nor Hutton explicitly discloses the first digital voice data is "not transmission when representative of audio signals below a predetermined threshold level". However, Reimer discloses the first digital voice data is not packetized for transmission when representative of audio

signals below a predetermined threshold level (see FIG. 5, Steeps 52,54,55,58; speech digital data is not framed/packetized for transmission by waiting when speech signal is lower than predetermined threshold; see 6, line 10-32). Therefore, it would have been obvious to one having ordinary skill in the art at the time the invention was made to provide "not transmission when representative of audio signals below a predetermined threshold level" as taught by Reimer, in the combined system of Berken and Hutton, so that it would provide capability to detect non-zero zero-crossing frames as suggested by Reimer; see Reimer col. 6, line 10-30.

Regarding Claims 61 and 62, Hutton discloses wherein the wireless packet network uses an Internet Protocol (IP), wherein IP protocol is TCP/IP (see col. 3, line 55-60; col. 2, line 60-67; see col. 5, line 1-10; utilizing TCP/IP in wireless network). Therefore, it would have been obvious to one having ordinary skill in the art at the time the invention was made to provide TCP/IP, as taught by Hutton in the system of Berken, so that it would provide exchanging realtime voice/video IP packet with IP address between two end units via Internet; see Hutton col. 1, line *50-65.*

Regarding Claim 68, Berken disclose a frequency hopping spread spectrum technique (see page 11, line 20-31; frequency hoping system of spread spectrum coding).

Regarding Claim 69, Berken disclose a direct sequence spread spectrum technique (see page 11, line 20-31; direct sequence spread spectrum coding).

Regarding Claim 70, Berken disclose wherein the at least one processor (see FIG. 1A, C; a combined system of processor 215, switch 213, phone 209) is further

operable to cause routing of digital voice data packets over a wired network (see page 9, line 1-10; see page 10, line 17-30; the combined system of 215, 213 and 209 routes/forwards voice packets over PSTN, Ethernet LAN, or Token Ring LAN).

Regarding Claim 71, Berken disclose wherein the routing of a call is selected by the first user (see FIG. 1A,C; a user enters/selects (from user input terminals 169,165 or 127) destination address/number (i.e. the routing of a call) in order to establish the call/connection; see page 9, line 1-10; see page 10, line 17-30).

Regarding Claim 72, Berken disclose the wired network comprises a packet network (see FIG. 1A, see page 9, line 1-10; see page 10, line 17-30; Ethernet LAN, or Token Ring LAN).

Regarding Claims 73, Berken discloses the wired network is a conventional switched telephone network (see FIG. 1A, PSTN 151; see page 9, line 1-10; see page 10, line 17-30).

10. Claims 63-65 are rejected under 35 U.S.C. 103(a) as being unpatentable over Berken in view of Hutton, and further in view of Lewen (US005341374A).

Regarding Claim 63, the combine system of Berken and Hutton discloses wherein the at least one processor received digital voice data and conversion of digital voice data as set forth above in claim 60.

Neither Berken nor Hutton explicitly discloses queues received data and delays conversion of queued data for an adjustable period of time.

However, Lewen teaches queuing (see FIG. 4, queuing/storing/collecting common memory 80) received digital voice data (see FIG. 2, collect received samples 120; see col. 14, line 44-49) and delays conversion of queued digital voice data for an adjustable period of time (see FIG. 2, delay time for storing/collecting voice samples in the memory before packetizing is adjusted between Tw (walktime) up to Tbfr (buffer storage time)); see col. 15, line 5-9,15-30. Therefore, it would have been obvious to one having ordinary skill in the art at the time the invention was made to queue received data and delays conversion of queued data for an adjustable period of time, as taught by Lewen in the combined system of Berken and Hutton, so that it would provision a communication system which effectively provides integrated voice, data and video communication and also provide real time reception of voice communication; see Lewen col. 2, line 50-62.

Regarding Claim 64, Lewen further discloses adjusts the period of time based upon a network propagation delay (see col. 13, line 56-66; see col. 14, line 22-39; see col. 15, line 5-9,15-30; adjusting delay time according Tw (propagation delay)). Therefore, it would have been obvious to one having ordinary skill in the art at the time the invention was made to adjust the period of time based upon a network propagation delay, as taught by Lewen in the combined system of Berken and Hutton, for the same motivation as set forth above in claim 63.

Regarding Claim 65, Lewen further discloses adjustable period of time using a packet sent to the communication device in response to a packet sent by the communication device (see col. 13, line 56-66; see col. 14, line 22-39; see col. *15,* line

5-9, 15-30; adjusting delay time according Tw (propagation delay), which is a time required for a signal bit of a frame/packet to travel from transmitting node to receive node). Therefore, it would have been obvious to one having ordinary skill in the art at the time the invention was made to provide adjustable period of time using a packet sent to the communication device in response to a packet sent by the communication device, as taught by Lewen in the combined system of Berken and Hutton, for the same motivation as set forth above in claim 63.

11. Claim 66 is rejected under 35 U.S.C. 103(a) as being unpatentable over Berken in view of Hutton and Lewen, and further in view of McKee (US005477531A).

Regarding Claim 66, neither Berken, Hutton nor Lewen explicitly disclose a test packet. However, McKee discloses determining propagation delay or queuing delay by utilizing in response to test packet sent by the communication device (see FIG. 2, test packet; see col. 1, line 60 to col. 2, line 60). Therefore, it would have been obvious to one having ordinary skill in the art at the time the invention was made to provide a test packet, as taught by McKee, in the combined system of Berken, Hutton and Lewen, so that it would provide to determine/test propagation delay or queuing delay; see McKee abstract col. 2, line 20-32.

12. Claim 67 is rejected under 35 U.S.C. 103(a) as being unpatentable over Berken in view of Hutton, and further in view of Cripps (US005838730A).

Regarding Claim 67, Berken disclose a frequency hopping spread spectrum technique (see page 11, line 20-31; frequency hoping system of spread spectrum coding). Berken does not explicitly disclose a frequency of approximately 2.4 gigahertz. However, using 2.4 GHz frequency hopping is well known in the art as defined by FCC. In particular, Cripps discloses wherein the wireless packet network communicates at a frequency of approximately 2.4 gigahertz (abstract; see col. 2, line 13-20; see col. 36, line 32-45; 2.4 GHz). Therefore, it would have been obvious to one having ordinary skill in the art at the time the invention was made to provide 2.4 GHz, as taught by Cripps, in the combined system of Berken and Hutton, so that it would provide a transmitter/receiver in accordance with FCC rules for 2.4 GHz ISM which is low cost and low power; see Cripps col. 2, line 15-32.

13. Claim 81 is rejected under 35 U.S.C. 103(a) as being unpatentable over Berken in view of Hutton, and further in view Dezonno (US 5991394).

For claim 81, Berken and Hutton fails to explicitly disclose a user is "*prompted*". However, Dezonno discloses a user (see FIG. 1, Agent 104, or user 102) is prompted to select delivery of voice (see FIG. 1, 2, prompt to select voice telephone call) to the called party (see FIG. 1, to user 102, or agent 104) by <u>one</u> of the circuit switched network (see FIG. 1, by telephone network 122) and the packet-based network (see FIG. 1, by Internet 108), if such prompting is indicated by a user defined parameter (see FIG. 1, when prompting is indicated by user's parameter such as telephone number or

name; see col. 3, line 33 to col. 4, line 25; see col. 4, line 40 to col. 5, lines 65; col. 7, line 25-32).

Therefore, it would have been obvious to one having ordinary skill in the art at the time the invention was made to provide a user is "*prompted*" as taught by Dezonno, in the combined system of Berken, Richter, Bonnaure et al. and Weaver, so that it would provide establishing voice communications between a computer user and an agent of a business, and a computer user to easily and conveniently have a business advertising on a computer network, call the computer user back over the telephone; see Dezonno col. 2, line 10-14, col. 3, lines 1-5.

Conclusion

Any inquiry concerning this communication or earlier communications from the examiner should be directed to MOHAMMAD ANWAR whose telephone number is (571)270-5641. The examiner can normally be reached on Monday-Thursday, 9am-4pm.

If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, Derrick W. Ferris can be reached on 571-272-3123. The fax phone number for the organization where this application or proceeding is assigned is 571-273-8300.

Information regarding the status of an application may be obtained from the Patent Application Information Retrieval (PAIR) system. Status information for published applications may be obtained from either Private PAIR or Public PAIR. Status information for unpublished applications is available through Private PAIR only. For more information about the PAIR system, see http://pair-direct.uspto.gov. Should you have questions on access to the Private PAIR system, contact the Electronic Business Center (EBC) at 866-217-9197 (toll-free). If you would like assistance from a USPTO Customer Service Representative or access to the automated information system, call 800-786-9199 (IN USA OR CANADA) or 571-272-1000.

MOHAMMAD ANWAR
Examiner
Art Unit 2463

/M. A./
Examiner, Art Unit 2463

/Derrick W Ferris/
Supervisory Patent Examiner, Art Unit 2463